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EXAMINER	
CLARK, MAXWELL A	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/510,222	KLEINER, PATRICK	
	Examiner	Art Unit	
	MAXWELL A. CLARK	2416	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *29 August 2008*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *14-31* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *14-31* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *10/04*.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments with respect to claims 14-31 have been considered but are moot in view of the new ground(s) of rejection.

Claim Rejections - 35 USC § 112

1. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

2. Claim 31 is rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Regarding claim 31, the phrase "such that" renders the claim indefinite because it is unclear whether the limitations following the phrase are part of the claimed invention. See MPEP § 2173.05(d).

Claim Rejections - 35 USC § 102

3. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

Claims 14-24, 26 and 30 are rejected under 35 U.S.C. 102(e) as being anticipated by Bradd et al. (US 203/0118002).

Regarding claim 14, Bradd discloses an IP/IP-gateway for processing signaling data and for controlling a connection of a voice communication link between at least two communication devices assigned to different IP-networks or different domains within IP-network (see figure 1; see figure 2; see figure 4, which illustrates the same as figure 2 of the instant application), the network IP/IP-gateway comprising: a signaling transmission unit for converting the signaling data format of signaling data originating from a first domain into a data format suitable for forwarding the signaling data to a second domain (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information

for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12); and a media transmission unit for converting the media data format of payload data originating from the first domain and associated with the voice communication link into a data format suitable for forwarding the payload data to the second domain (see figure 1-6,14, 12; see paragraph 0028 media gateway 6 wishes to initiate a VoIP call with media gateway 12 to provide a telephony connection between telephone terminal 20 (connected to media gateway 6) and telephone terminal 22 (connected to media gateway 12); see figure 2-6,26,24,28,12; see paragraph 0033, the address translator 24 is able to modify addresses in the IP packets it receives (both in the payload and header depending on protocol used) so that media gateway 6 views virtual gateway 26 as its destination terminal and media gateway 12 similarly, views virtual gateway 28 as its destination terminal. Data is routed through the translator between the virtual gateways 26 and 28 so that a path is established between the two media gateways 6 and 12 having identical IP addresses; see figure 4-6, 26, 24', 28',

26", 24", 28", 12; see paragraph 0045, in protocol exchange 44, call server 18" provides information to call server 16" about the SDP information for public virtual gateway 26" on address translator 24". In protocol exchange 46, call server 16" provides information to the address translator 24' about the mapping which it should make between virtual gateway 24" on address translator 26" and gateway 12 and finally in protocol exchange 48, gateway 6 begins to send RTP packets to address translator 24'), wherein the signaling transmission unit comprises further communication mechanisms for controlling the media transmission unit using the signaling data (see figure 1-NCS(SDP); see figures 2-Device Control; see paragraph 0032 the call server, i.e. signaling transmission unit, is able to communicate with the media gateway, i.e. media transmission unit; see paragraph 0037, in protocol exchange 38, call server 18 provides call server 16' with SDP information for the gateway 12).

Regarding claim 15, Bradd discloses the signaling transmission unit controls the media transmission unit according to a master/slave relationship (see paragraph 0042, the call server, i.e. signaling transmission unit, directs data on its external virtual gateway 28' to the external gateway 26" of address translator 24," i.e. master/slave relationship).

Regarding claim 16, Bradd discloses the master/slave relationship comprises determination of the status, and/or capacity utilization, and/or functionality of the respective media transmission unit (see paragraph 0042, the call server directs data on its external virtual gateway 28' to the external gateway 26" of address translator 24,

corresponds to the determination of functionality of the media transmission unit, i.e. the gateway).

Regarding claim 17, Bradd discloses the signaling transmission unit comprises a communication mechanism for converting a network address format of signaling data originating from a first domain into a network address format suitable for forwarding the signaling data to a second domain (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can

recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 18, Bradd discloses the signaling transmission unit comprises a communication mechanism for converting a network address format of signaling data originating from a first domain into a network address format suitable for forwarding the signaling data to a second domain (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway

26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 19, Bradd discloses the signaling transmission unit comprises a communication mechanism for converting a network address format of signaling data originating from a first domain into a network address format suitable for forwarding the signaling data to a second domain (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in

protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 20, Bradd discloses the signaling transmission unit comprises a communication mechanism for terminating signaling data originating from a first domain

and relating to performance features that are valid in the first domain (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing

to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 21, Bradd discloses the signaling transmission unit comprises a communication mechanism for terminating signaling data originating from a first domain and relating to performance features that are valid in the first domain (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30,32,34',36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route

from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 22, Bradd discloses the signaling transmission unit comprises a communication mechanism for terminating signaling data originating from a first domain and relating to performance features that are valid in the first domain (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send

packets for this call to media gateway 12; see figure 4-16”, 18”; see figure 5-30,32,34’,36’; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 23, Bradd discloses the signaling transmission unit comprises a communication mechanism for terminating signaling data originating from a first domain and relating to performance features that are valid in the first domain.

Regarding claim 24, Bradd discloses the signaling transmission unit comprises a communication mechanism having a firewall proxy functionality for enabling the payload data associated with the voice connection to pass a data firewall (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16’, 18; see figure 3-30, 32 34' 36'; see paragraph 0034,

after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16”, 18”; see figure 5-30,32,34’,36’; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12).

Regarding claim 26, Bradd discloses the signaling transmission unit comprises a communication mechanism for at least one of converting and monitoring and blocking performance features (see paragraph 0038, in protocol exchange 40, call server 16' then (knowing that the call will be routed via the address translator 24), tells the address translator 24 about the correct mapping to make between virtual gateway 28 and the media gateway 12, i.e. converting performance features).

Regarding claim 30, Bradd discloses wherein a transmission unit comprises one of the signaling transmission units and one of the media transmission units provided on separate hardware platforms (see figures 1, 2 and 4, signaling transmission units 16 and 18 are illustrated on separate hardware platforms as are the media transmission units 6 and 12).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. The factual inquiries set forth in *Graham* **v.** *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

Claims 25 and 28 are rejected under 35 U.S.C. 103(a) as being unpatentable over Bradd et al. (US 203/0118002) in view of Li et al. (USPN 6,591,301 B1).

Bradd discloses the claimed limitations in paragraph 3 above. Bradd does not expressly disclose the following: regarding claim 25, a communication mechanism for controlling the volume of traffic and for preventing overload; regarding claim 28, a communication mechanism for controlling the volume of traffic and for preventing overload. Li discloses controlling network gatekeeper message processing include overload control routines executed by a network gatekeeper among other devices comprising the following features: regarding claim 25, a communication mechanism for controlling the volume of traffic and for preventing overload (col. 4, lines 17-19, wherein regulating incoming message traffic in manner structured to favor calls in progress over newly originating calls corresponds to a communication mechanism for controlling the volume of traffic and for preventing overload; col. 14, lines 18-34, wherein the overload control is for H.323, i.e. signaling transmission unit, and Media Gateway control, i.e. media transmission unit); regarding claim 28, a communication mechanism for controlling the volume of traffic and for preventing overload (col. 4, lines 17-19, wherein regulating incoming message traffic in manner structured to favor calls in progress over newly originating calls corresponds to a communication mechanism for controlling the volume of traffic and for preventing overload; col. 14, lines 18-34, wherein the overload control is for H.323, i.e. signaling transmission unit, and Media Gateway control, i.e. media transmission unit) for the purpose reducing the prospect of failure. It would have

been obvious to one of ordinary skill in the art at the time of the invention to include the teaching of Li in Bradd for the purpose of regulating incoming message traffic to prevent overload conditions so to prevent crashing during overload conditions, see abstract.

Claim 27 is rejected under 35 U.S.C. 103(a) as being unpatentable over Bradd et al. (US 203/0118002) in view of MeLampy et al. (US 2003/0016664 A1).

Bradd discloses the claimed limitations in paragraph 3 above. Bradd does not expressly disclose converting priority identifiers of signaling data originating from a first domain into priority identifiers suitable for forwarding the signaling data to a second domain. MeLamphy discloses providing rapid rerouting of real-time transport protocol (RTP) multi-media flows among multiple devices featuring the following: regarding claim 27, MeLamphy discloses a communication mechanism for converting priority identifiers of signaling data originating from a first domain into priority identifiers suitable for forwarding the signaling data to a second domain (¶0038, wherein once a forwarding decision is made, i.e. forward into a second domain, the traffic manager queues the received packet into its respective IP flow and associated priority which corresponds to a communication mechanism for converting priority identifiers of signaling data originating from a first domain into priority identifiers suitable for forwarding the signaling data to a second domain) for the purpose managing packets according to priority. It would have been obvious to one of ordinary skill in the art at the time of the application to include the teachings of MeLampy in Bradd for the purpose of enforcing IP session data flow rates, see paragraph 0038.

Claim 29 is rejected under 35 U.S.C. 103(a) as being unpatentable over Bradd et al. (US 203/0118002) in view of Chu et al. (US 7,200,139 B1).

Bradd discloses the claimed limitations in paragraph 3 above. Bradd does not expressly disclose transmission unit comprises one of the signaling transmission units and one of the media transmission units provided on a common hardware platform. Chu discloses a Software Radio Port (SRP) which functions as a VoIP gateway among other features including the following: regarding claim 29, Chu discloses transmission unit comprises one of the signaling transmission units and one of the media transmission units provided on a common hardware platform (see figure 2-SRP; column 3, lines 43-45, SRP may be further equipped with VoIP capability with a VoIP Media Gateway 205 and VoIP Signaling Gateway) in order to manage VOIP call processing. It would have been obvious to one of ordinary skill in the art at the time of the present application to include the teachings of Chu in Bradd so that signaling and media transmission, i.e. Media Gateway and Signaling Gateway 207, can be integrated and interface an IP/Ethernet interface, see column 3, lines 44-47.

Claim 31 is rejected under 35 U.S.C. 103(a) as being unpatentable over Bradd et al. (US 203/0118002) in view of Nodoushani (US 6,563,816 B1).

Bradd discloses signaling data and for controlling a connection of a voice communication link between at least two communication devices assigned to different packet-switched communication networks or different domains within a communication network, such that voice data packets are forwarded directly from one of the different packet-switched communication networks or different domains to at least another one of

the different packet-switched communication network or different domains comprising (see figure 1; see figure 2; see figure 4, which illustrates processing signaling data and for controlling a connection of a voice communication link between at least two communication devices assigned to different domains within a communication network, such that voice data packets are forwarded directly from one of the different packet-switched communication networks or different domains to at least another one of the different packet-switched communication network or different domains comprising): converting a data format of signaling data originating from a first domain into a data format suitable for forwarding the signaling data to a second domain (see figure 1-16, 18; see paragraph 0027, call servers 16 and 18 which are used to set up calls (using the SIP telephony protocol) carrying some ISUP variant and session descriptor protocol (SOP) information; see figure 2-16', 18; see figure 3-30, 32 34' 36'; see paragraph 0034, after the initial call set-up exchanges between the gateway 6 and the call server 16', the call server 16' (in protocol exchange 30) determines, based on translations, that the call must go out on a SIP-T route. The particular SIP-T route is provisioned to indicate that the call goes through the address translator 24. In protocol exchange 32, the call server 16' tells the address translator 24 about the SDP information for media gateway 6; see figure 4; paragraph 0044, protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12; see figure 4-16", 18"; see figure 5-30, 32, 34',

36'; paragraph 0044, In protocol exchange 36' call server 16" carries out a typical SIP-T exchange with call server 18" but using the SDP Information for public virtual gateway 28'. Call server 18" needs to know that the incoming call from call server 16" is from another domain. It can recognize this based on provisioning of the route from call server 16" being in another IP address space. It needs to know this so it can get its address translator 24" into the call. In protocol exchange 37, call server 2 informs address translator 24" of the public information for address translator 24'. Then in protocol exchange 38', address translator 24" provides call server 18" with information about virtual gateway 28' for passing to gateway 12 and virtual gateway 26" for passing to call server 16" to pass on to address translator 24'. Then in protocol exchange 40', call server 18" tells address translator 24" to send packets for this call to media gateway 12); converting the data format of payload data originating from a first domain and associated with the voice communication link into a data format suitable for forwarding the payload data to a second domain (see figure 1-6,14, 12; see paragraph 0028 media gateway 6 wishes to initiate a VoIP call with media gateway 12 to provide a telephony connection between telephone terminal 20 (connected to media gateway 6) and telephone terminal 22 (connected to media gateway 12); see figure 2-6,26,24,28,12; see paragraph 0033, the address translator 24 is able to modify addresses in the IP packets it receives (both in the payload and header depending on protocol used) so that media gateway 6 views virtual gateway 26 as its destination terminal and media gateway 12 similarly, views virtual gateway 28 as its destination terminal. Data is routed through the translator between the virtual gateways 26 and 28 so that a path is

established between the two media gateways 6 and 12 having identical IP addresses; see figure 4-6, 26, 24', 28', 26", 24", 28", 12; see paragraph 0045, in protocol exchange 44, call server 18" provides information to call server 16" about the SDP information for public virtual gateway 26" on address translator 24". In protocol exchange 46, call server 16" provides information to the address translator 24' about the mapping which it should make between virtual gateway 24" on address translator 26" and gateway 12 and finally in protocol exchange 48, gateway 6 begins to send RTP packets to address translator 24'); and forwarding the converted signaling data and payload data to the second domain (see paragraphs 0038-0039, in protocol exchange 40, call server 16' then (knowing that the call will be routed via the address translator 24), tells the address translator 24 about the correct mapping to make between virtual gateway 28 and the media gateway 12. Finally, in protocol exchange 42, gateway 6 is instructed to send real time protocol (RTP) packets to the address translator 24. It will be noted, that the address translator 24 is configured automatically to allocate a UDP port for real time control protocol (RTCP) which is equal to the RTP UDP port (typically an even number) plus 1. Data then flows between the gateways 6 and 12 (via address translator 24) in the conventional manner). Bradd does not expressly disclose the signaling data and payload data are synchronized by a control system using the signaling data. Nodoushani discloses using a first signaling protocol such as Media Gateway Control Protocol, Session Initiation Protocol or H.323, and with the local digital switch using a second signaling protocol such as GR-303 for controlling call processing in the communication system among the following features: regarding claim 31, Nodoushani

discloses that the signaling data and payload data are synchronized by a control system using the signaling data (see column 15, lines 57-59, controller 100 performs timing on the signal received, signaling data, to keep the voice, i.e. payload data, synchronized). It would have been obvious to one of ordinary skill in the art at the time of the present application to include the teachings of Nodoushani in Bradd for the so that the signaling and payload data timing coincide in time and for controlling call processing in the communication system among the following, see abstract.

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Barany, Peter A. et al. (US 20020034166 A1); Donovan; Steven R. et al. (US 6453034 B1); Nodoushani; Paiman et al. (US 6563816 B1); Foti; George (US 6963583 B1); Elliott; Isaac K. et al. (US 6614781 B1); Bradd, Patrick et al. (US 20030118002 A1).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to MAXWELL A. CLARK whose telephone number is (571) 270-1956. The examiner can normally be reached on Monday to Thursday 7:30A.M. through 5:00P.M. Eastern Standard Time.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Yao B. Kwang can be reached on (571) 272-3182. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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